(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number: **0 132 216**
A1

(12) EUROPEAN PATENT APPLICATION

(21) Application number: 84630096.0

(22) Date of filing: 15.06.84

(51) Int. Cl.⁴: G 10 L 7/06

(30) Priority: 17.06.83 AU 9872/83

(43) Date of publication of application: 23.01.85 Bulletin 85/4

(84) Designated Contracting States: AT BE CH DE FR GB IT LI LU NL SE

(71) Applicant: THE UNIVERSITY OF MELBOURNE
Grattan Street
Parkville Victoria 3052(AU)

(72) Inventor: Dewhurst, David John
37 Cullwell Avenue
Mitcham 3132 Victoria(AU)

(72) Inventor: Ng, Chee Wei
10 JN, Union 3
Sentul Kuala Lumpur 13-06(MY)

(72) Inventor: Hughes, Murray Allan
25 Gwynn Street
Mount Waverley 3149 Victoria(AU)

(72) Inventor: Johnson, Donald Archibald Harley
Lizbeth Avenue
Mitcham 3132 Victoria(AU)

(74) Representative: Waxweiler, Jean et al,
OFFICE DENNEMEYER S.à.r.l. 21-25 Allée Scheffer
P.O.Box 41
L-2010 Luxembourg(LU)

(54) Signal processing.

(57) The disclosed system for extracting desired information from a speech signal includes means for taking overlapping samples of an utterance, computer means programmed to test each sample to determine whether it is voiced or unvoiced and for performing the following operations on each voiced sample:

applying a 30 ms. Hamming window to smooth the edge of the signal and to ensure that false artifacts will not be present in the following processing stage, obtaining a magnitude spectrum using at least 1024 points Fast Fourier transform, obtaining the log of the magnitude spectrum, compressing the spectrum, performing a three-point filter algorithm a suitable number of times, expanding the spectrum so obtained and locating the dominant peaks in the resulting spectrum to give the information content contained in said speech signal. The specification also discloses the time equivalent of the above method.

EP 0 132 216 A

./...

AL

Croydon Printing Company Ltd

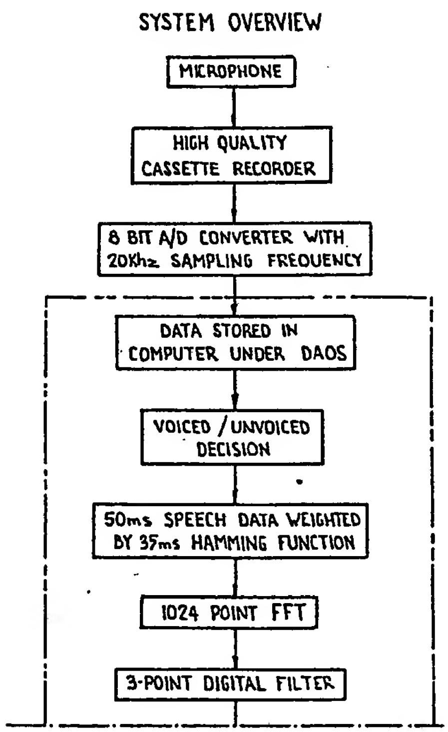

SYSTEM OVERVIEW
MICROPHONE
HIGH QUALITY CASSETTE RECORDER
8 BIT A/D CONVERTER WITH 20Khz SAMPLING FREQUENCY
DATA STORED IN COMPUTER UNDER DAOS
VOICED / UNVOICED DECISION
50ms SPEECH DATA WEIGHTED BY 35ms HAMMING FUNCTION
1024 POINT FFT
3-POINT DIGITAL FILTER

FIG. 7A.

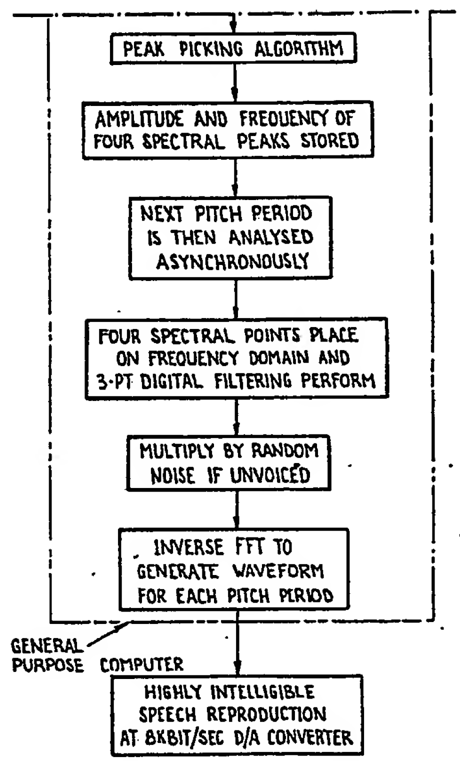

PEAK PICKING ALGORITHM
AMPLITUDE AND FREQUENCY OF FOUR SPECTRAL PEAKS STORED
NEXT PITCH PERIOD IS THEN ANALYSED ASYNCHRONOUSLY
FOUR SPECTRAL POINTS PLACE ON FREQUENCY DOMAIN AND 3-PT DIGITAL FILTERING PERFORM
MULTIPLY BY RANDOM NOISE IF UNVOICED
INVERSE FFT TO GENERATE WAVEFORM FOR EACH PITCH PERIOD
GENERAL PURPOSE COMPUTER
HIGHLY INTELLIGIBLE SPEECH REPRODUCTION AT 8KBIT/SEC D/A CONVERTER

FIG. 7B.

TITLE: SIGNAL PROCESSING.

BACKGROUND OF THE INVENTION

This invention relates to a system for the processing of signals to extract desired information. The invention is particularly applicable to the extraction of the desired information content from a received speech signal for subsequent use in activating or stimulating an implantable hearing prosthesis or for other purposes.

The variability of speech signals between speakers of the same utterance (as shown in Figure 1) has been a major problem faced by all speech scientists. However, the fact that human auditory system is capable of extracting relevant speech information from widely varying speech signals has baffled speech researchers for decades. The information must of course be present in the signal but thus far researchers in this field have been unable to devise a system for reliably extracting the information from a speech signal.

The retrieval of text from voice involving recognition of unrestricted speech is still considered to be far beyond the current state of the art. What is being attempted is automatic recognition of words from restricted speech. Even so, the reliability of these ASR (Automatic Speech Recognition) systems is unpredictable. One report ("Selected Military Applications of ASR Technology" by Woodward J.P. & Cupper E.J., IEEE Communications Magazine, 21, 9

December 1983, pg 35-41) lists eighty different factors which can affect their reliability. Such advances in ASR as have been achieved have arisen more from improved electronics and microprocessor chips than from the development of any new technology for ASR.

In considering this question, the present inventors have given consideration to the manner in which the auditory system handles widely varying speech signals and extracts the information required to make the speech signal intelligible. When sounds of speech are transmitted to the higher centres of the brain by means of the auditory system it undergoes several physiological processes.

When speech signals arrive at the middle ear, a mechanical gain control mechanism acts as an automatic gain control function to limit the dynamic range of the signal being analysed. According to the temporal-place representation, the discharge patterns of auditory nerve-fibres show stronger phase locking behaviour to spectral peaks than locking to other harmonics of the stimulus. At physiological sound level, synchrony to dominant spectral peaks saturates and responses to pitch harmonics are suppressed. The resulting effect is such that the rough appearance of the pitch harmonics are masked out.

SUMMARY OF THE INVENTION

The present inventors therefore determined that if the pitch information (that is, the speaker attributes, which contain no real information) such as pitch frequency, harmonics components thereof and other minor speaker attributes, could be removed from the speech signal, the remaining signal would contain the information necessary for understanding the utterance contained in the complex speech signal thereby

resulting in a signal usable to stimulate a hearing prosthesis or for other purposes, such as, speech recognition by computer, speech synthesis, and speech bandwidth compression for rapid transmission of speech.

In its broadest aspect therefor, the invention provides a system for extracting desired information from a speech signal including means for performing the essential steps of removing from or suppressing in the speech signal at least the significant components relating to pitch frequency, and identifying and tracking in time the spectral peaks of the resulting signal.

BRIEF DESCRIPTION OF THE DRAWINGS

In the drawings:

Figure 1 is a plot showing the variability in the speech signals of two different speakers of the same utterance;

Figures 2 and 3 are spectral plots of frequency against time of the utterance shown in Figure 1 again showing the variability of the signals;

Figure 4 shows the effect of applying a smoothing algorithm to the signal;

Figures 5 and 6 show plots of the spectral peaks produced by the smoothing shown in Figure 4;

Figure 7 is a schematic representation of one signal processing method embodying the invention;

Figures 8 to 15 show the steps in the processing method as applied to a specific utterance;

Figure 15A is a three-dimensional plot of the spectral peak variation against time of the utterance 'Boat';

Figure 16 is a schematic representation of a real-time speech synthesizer;

Figures 17 and 17A show typical line representations of the utterance 'Melbourne' to be used in the synthesizer described in Figure 16; and

Figures 18 to 23 show the steps in an alternative processing method.

DESCRIPTION OF PREFERRED EMBODIMENTS

The aim of many techniques of analysing speech signals is to characterize the temporal variation of the amplitude spectra of short intervals of a word. A digital method of producing a frequency spectrum of a short time interval by means of the Fast Fourier Transform (FFT) will yield a "messy" spectrum caused by pitch harmonics. Plots of these spectral variations against time shown in Figures 2 and 3 will be seen to be masked by the dominant pitch harmonics.

A smoothing algorithm is performed on the signal "noise" in the spectrum and is filtered and the centre frequency and amplitude of the four locally dominant spectral peaks are able to be picked out (see Figure 4). Plots of these spectral peaks against time are shown in Figures 5 and 6. The similarities in these plots between speakers are clearly evident particularly in the direction of movement of the spectral peak tracks. Unlike formants, these spectral lines are discontinuous and their movements cover a wider bandwidth. There is little doubt that this

concept of processing is the first step towards speech perception.

Using the information acquired by the above process, a reverse processing technique can be used to resynthesize highly intelligible speech on a digital computer. The same information can be displayed in two dimensions as line patterns and by means of an optical reader these lines may be converted back into speech frequencies. Using this concept it can be demonstrated that intelligible speech can be produced on a real-time hardware synthesizer even without amplitude variations.

It is envisaged that this method of speech processing can offer data rate reduction of the order of 1:40 without subjectively losing much fidelity in speech transmission.

Various methods of achieving the above described ends may be applied to the speech signal and two different approaches will now be described in greater detail.

In the first processing approach, the signal is received and processed in the manner schematically outlined in Figure 7. The process begins with the sampling of a prefiltered speech signal at a rate of about 20000 samples per second. The sampled speech is then analyzed in segments of duration 50ms. Successive 50ms segments are analyzed at 10ms intervals so that there is an overlap of adjacent segments to provide the necessary continuity. The processing technique may be better understood by considering the following example of an actual speech signal conveying the word 'boat'. The process involves the following steps of:

(a) Taking a 50ms speech sample from the word BOAT("O"), (Fig. 8)

(b) Applying the voiced/unvoiced test (as described further below),

(c) Applying a 30ms Hamming window (Fig. 9) to smooth the edge of the signal and to ensure that false artifacts will not be present in the following processing stage,

(d) Obtaining a magnitude spectrum using at least 1024 points Fast Fourier Transform (Fig. 10),

(e) Log of the magnitude spectrum (Fig. 11),

(f) Spectrum compression (Fig. 12),

(g) Three-point filter algorithm is applied a suitable number of times, (Fig. 13),

(h) Spectrum is expanded as in (Fig. 14),

(i) Four dominant peaks are located as described in the mathematical details given below (Fig. 15).

Figure 15A shows the spectral peaks extracted by the above method in a three-dimensional plot.

When a 50ms segment is transformed by the discrete Fourier Transform process, the resulting spectrum consists of a number of spectral lines occurring at frequencies which are multiples of 20Hz. The distribution of amplitudes of these lines across the frequency range, however, indicates the true distribution of spectral energy of the speech segment. The human observer can pick out the peaks of the spectral energy (i.e. the positions where the energy distribution has obvious maxima) by eye with little difficulty (see Figs. 2 and 3). The above described technique enables a computer to perform a similar task but the process is quite involved especially as care has to be taken to eliminate artifacts of the sampling process which have nothing to do with the original speech segment. The process also smooths out other features of the

spectrum dependant on pitch pulse spectral energy, speaker specific characteristics and the like.

The discrete Fourier Transform is performed by the Fast Fourier Transform routine.

$$X_n(e^{j2\pi/N}) = \sum_{m=0}^{N-1} y(n-m)x(m)e^{-j2\pi m/N}$$

N=1024 points

y(n) is a suitable raised cosine window.

The three point filter algorithm is represented by:

$$p(n) = \left[\frac{p(n-1)}{4} + \frac{p(n)}{2} + \frac{p(n+1)}{4} \right]$$

For a function as shown below

$$X(k) = [\, x(k-1)/4 + x(k)/2 + x(k+1)/4 \,]$$

the corresponding time sequence would be

$$= W_N^{n}\, x(n)/4 + W_N^{-n}\, x(n)/4 + x(n)/2 \quad \text{where } W_N = e^{-j(2\pi/N)}$$

$$= e^{-2\pi jn/N}\, x(n)/4 + e^{2\pi jn/N}\, x(n)/4 + x(n)/2$$

$$= 1/2\, x(n) \quad [1 + \cos 2\pi n/N]$$

i.e.

$$X(k) <> 1/2x(n)\, [1+\cos(2\pi n/N)] \;\; = F_1\, [X(k)]$$

$$F_m(X(k)) = x(n)\, [1 + \cos\ (2\pi n/N)]^m$$

Thus the time domain equivalent of a three-point filtering on the frequency domain is multiplication by

$$x(n)\, [1 + \cos\ (2\pi n/N)]$$

Frequency compression on the magnitude spectrum is represented by:

$$p(n) = p(3n) \qquad \text{where } n = 1 \text{ to } 350$$

1024 points are compressed to 350 points by sampling

every third point.

The second derivative peak picking algorithm is represented by:

$$dy/dx = [p(n)-p(n-1)] = 0 = p'(n)$$

$$p''(n) = [p'(n)-p'(n-1)] = \text{negative}$$

When both these conditions are met the location of the peak is noted. A maximum of seven peaks can be located in the spectrum but only the four largest are selected.

A speech signal may be regarded as

$$\text{VOICED when } Ls = 1/M \sum_{N=N}^{N+M-1} a(n) \quad \text{is large}$$

$$\text{and as UNVOICED when } Ls = 1/M \sum_{n=M}^{N+M-1} a(n) \quad \text{is small}$$

$$\text{AND } Ld = 1/M \sum_{n=N}^{N+M-1} a(n+1)-a(n) \text{ is significant}$$

where Ls = absolute average level of 30ms of speech
Ld = absolute average level of 30ms of the differenced signal.

A voiced/unvoiced decision is made depending on the nature of the source of excitation of sounds. A voiced sound is perceived when the glottis is vibrating at a certain pitch causing pulses of air to excite the natural resonating cavities of the vocal tract. Unvoiced sounds are produced by a turbulent flow of air caused by a constriction at some point in the vocal tract. In analysing speech a decision is required to distinguish these so that a correct source of excitation can be used during synthesis. An algorithm can be written to define a voiced speech when the absolute

average signal is high and unvoiced when it is rapidly varying and of a small amplitude. If a signal sample is determined to be unvoiced it is disregarded in the analysis process.

The method employed limits the spectral peak resolution of the resulting spectrum. However, it is found that the centre frequency and the amplitude of four locally dominant spectral peaks are sufficient information for the auditory system to characterise the short term acoustic properties that distinguish one speech sound from another.

It is also known that auditory neural activities adapt themselves (neural adaption) whereby a high intensity stimulus will quickly reach saturation level. A similar process of adaptive frequency equalization is done on the frequency spectrum by transforming it to a log scale to ensure that the more important higher frequency components are not lost while keeping the stronger low frequency components within dynamic range. Furthermore, only the magnitude spectrum need be considered, since the cochlea is unable to resolve signal phase components.

A property of the cochlear and neural system is that it can only respond to changes of a time constant of the order of 10 ms. It is thus necessary that the processing technique employed extracts and updates its information rate every 10 ms.

Using the above method of processing, the information extracted, that is, the time variation of the spectral peaks movements, can be used as inputs to an implantable hearing prosthesis (such as described in Australian specifications AU-A 41061/78 and AU-A 59812/80 to mimic the function of the cochlea.

The same information can be used for speech recognition as illustrated in spectral plots

against time. Thirdly, using the information acquired, a reverse processing technique can resynthesize intelligible speech either on a digital computer or on a real-time hardware synthesizer.

During resynthesis, each spectral peak position is relocated in the frequency domain, without regard to phase. Three-point digital smoothing is done on these points to spread the spectrum. This would produce a decaying waveform for every pitch period generated in the time domain. The inverse FFT is performed and a data length corresponding to a pitch period is extracted.

For unvoiced speech, the spectrum is multiplied by a random phase function prior to inverse FFT. A 600Hz bandwidth for the noise spectral peak is satisfactory. The next set of data is decoded similarly until the end of the utterance.

In designing a real-time speech synthesizer as shown schematically in Figure 16, one must consider a method of converting these spectral lines into sine waves of frequencies from 0.3Khz. A linear 256-pixels RETICON chip is used. It is enclosed inside a commercial camera with focus and aperture size adjustments. The camera is mounted on an optical bench with a rotating drum at right-angles to it. Four controlled oscillators using X2206 function generator chips are required.

A start pulse every 10ms is used to start the count to locate the position of each line. A maximum of four lines may be identified, and the position of each line is decoded as an 8-bit address. The address is then latched, so that the D/A of each line is in continuous operation throughout the 10ms period. If the position of the line changesin the next 10ms, a 'new' address is latched. If the line disappears an analogue

switch will disable the oscillator.

The D/A comprises a ladder-network to allow up to 8-bits of accuracy in determining the current flow into the X2206 oscillator chip.

$$\text{Frequency} = 320\ I(\text{mA})/C(\mu F) \quad \text{Hz}$$

Having set a fixed capacitance, the frequency generated by the chip is only dependent of the position of the line. The output from the four oscillators are summed and multiplied by a triangular wave function with an offset. This procedure will generate a pitch period as well as spreading the spectrum wider as it appears in normal speech.

A typical line input representing the word 'Melbourne' is shown in Figure 17. In Figure 17A the base line has been removed since this does not contain any information and may be replaced by a straight line as shown. It has been established above that the variation with time of the frequencies at which the spectral energy maxima occur contains all the information necessary to resynthesize the spoken words. Moreover we have found that the changes in amplitude of the maxima are unimportant in resynthesizing understandable words (though they may be important in speaker identification) and the actual pitch frequency used is not critical at all. In this respect in particular the approach of this invention differs from that of others which endeavour to determine pitch frequency accurately. In the resynthesis process the outputs of three or four tone generators whose frequencies are controlled by the frequency peak 'tracks', are combined, and finally a tone representing the pitch frequency added in. This last step is not actually essential for intelligibility, but improves realism.

An alternative processing method, which can be shown to be mathematically "the time" equivalent of the above method, will now be briefly explained with reference to Figures 18 to 23. This processing method involves the following steps:

(a) A sample of the time waveform of the same utterance BOAT. (Fig. 18),
(b) Time expansion of the speech sample (Fig. 19),
(c) Applying a window of the form $(1 + \cos(\Pi t/T))^N$ (Fig. 20),
(d) Resulting waveform after windowing (Fig. 21),
(e) Obtaining a magnitude spectrum using at least 1024 points Fast Fourier transform,
(f) Log of the magnitude spectrum (Fig. 22), and
(g) Four dominant peaks are located (Fig. 23).

As in the case of the embodiment of Figure 7, each of the above operations are performed using a suitably programmed general purpose computer.

As mentioned above, other methods of achieving the same results may be easily devised using standard mathematical procedures. Similarly, the processing techniques by which the above described alternative processing steps may be performed in a computer will be well understood by persons skilled in the art and will not therefore be described in greater detail in this specification. The manner in which the extracted information is utilized will vary according to the application and although the processing technique was developed with application to a hearing prosthesis in mind, the technique clearly has wider application, several of which have been indicated above. Other applications include:

Control of plant and machines by spoken command.
Aids for handicapped - voice operated wheel chairs, voice operated word processors and braille writing systems.

Voice operation of computers.
Automatic information systems for public use activated by spoken commands.
Automatic typescript generation from speech.

CLAIMS:

1. A system for extracting desired information from a speech signal, including means (Fig. 7) for performing the essential steps of removing from or suppressing in the speech signal at least the significant components relating to pitch frequency, and identifying and tracking in time the spectral peaks of the resulting signal.

2. The system of claim 1, wherein said removal or suppression step comprises the steps of taking samples of the speech signal to be processed and filtering the samples to remove or suppress the pitch components therein whereby the locally dominant spectral peaks are more readily able to be located and tracked.

3. The system of claim 2, wherein the filtering of said signal is performed in accordance with a three point filter algorithm.

4. The system of claim 2 or 3, wherein said signal is Fourier transformed prior to said filtering.

5. The system of claim 4, wherein each signal component is tested to determine whether it is voiced or unvoiced and if unvoiced, said signal component is not subjected to Fourier transformation or filtering.

6. The system of claim 4 or 5, wherein a Hamming window is applied to each signal component before Fourier transformation to smooth the edge of the signal and to ensure that false artifacts will not be present in the following processing stage.

7. The system of claim 1, including means for performing the following steps:

(a) taking overlapping samples of said speech signal,

(b) testing each sample to determine whether the sample is voiced or unvoiced, and performing the following steps in connection with each voiced sample,

(c) applying a Hamming window to each sample,

(d) obtaining a magnitude spectrum by performing a Fast Fourier transform on each sample,

(e) obtaining the log of the magnitude spectrum of each sample,

(f) compressing the spectrum so obtained,

(g) performing a three-point filter algorithm on the compressed sample a plurality of times,

(h) expanding the spectrum so obtained, and

(i) locating the dominant peaks in said expanded spectrum.

8. The system of claim 2, wherein said filtering step comprises applying a low pass filtering function to each signal sample.

9. The system of claim 8 wherein said filtering function is of the form $(1 + \cos(\pi t/T))^N$

10. The system of claim 8 or 9 further including the step of Fourier transforming said component following filtering.

11. The system of claim 1, including means for performing the following steps:

(a) overlapping samples of the time waveform of the speech signal are taken,

(b) each sample is time expanded,

(c) a filtering function of the form $(1 + \cos(\pi t/T))^N$ is applied to each sample,

(d) the resulting signal is Fast Fourier transformed,

(e) the log of the resulting magnitude spectrum is obtained and,

(g) the dominant spectral peaks are located.

12. A system for synthesizing intelligible speech comprising means (Fig. 16) for storing a representation of said spectral peak information extracted by the system according to any preceding claim, and means (Fig. 16) for utilizing said spectral peak information to generate a synthesized utterance.

13. The system of claim 12 further comprising tone oscillator means having frequencies generally corresponding to each said spectral peak, means for varying the applied voltage producing each tone oscillation in accordance with the detected time variations in each spectral peak.

14. The system of claim 13 further comprising the addition of a tone representing pitch frequency to improve realism in the synthesized speech.

15. A method of extracting desired information from a speech signal comprising the steps of removing from or suppressing in the speech signal at least the significant components relating to pitch frequency, and identifying and tracking in time the spectral peaks of the resulting signal.

16. A method of synthesizing intelligible speech comprising the steps of storing a representation of said spectral peak information extracted according to the method of claim 15 and utilizing said spectral peak information to generate a synthesized utterance.

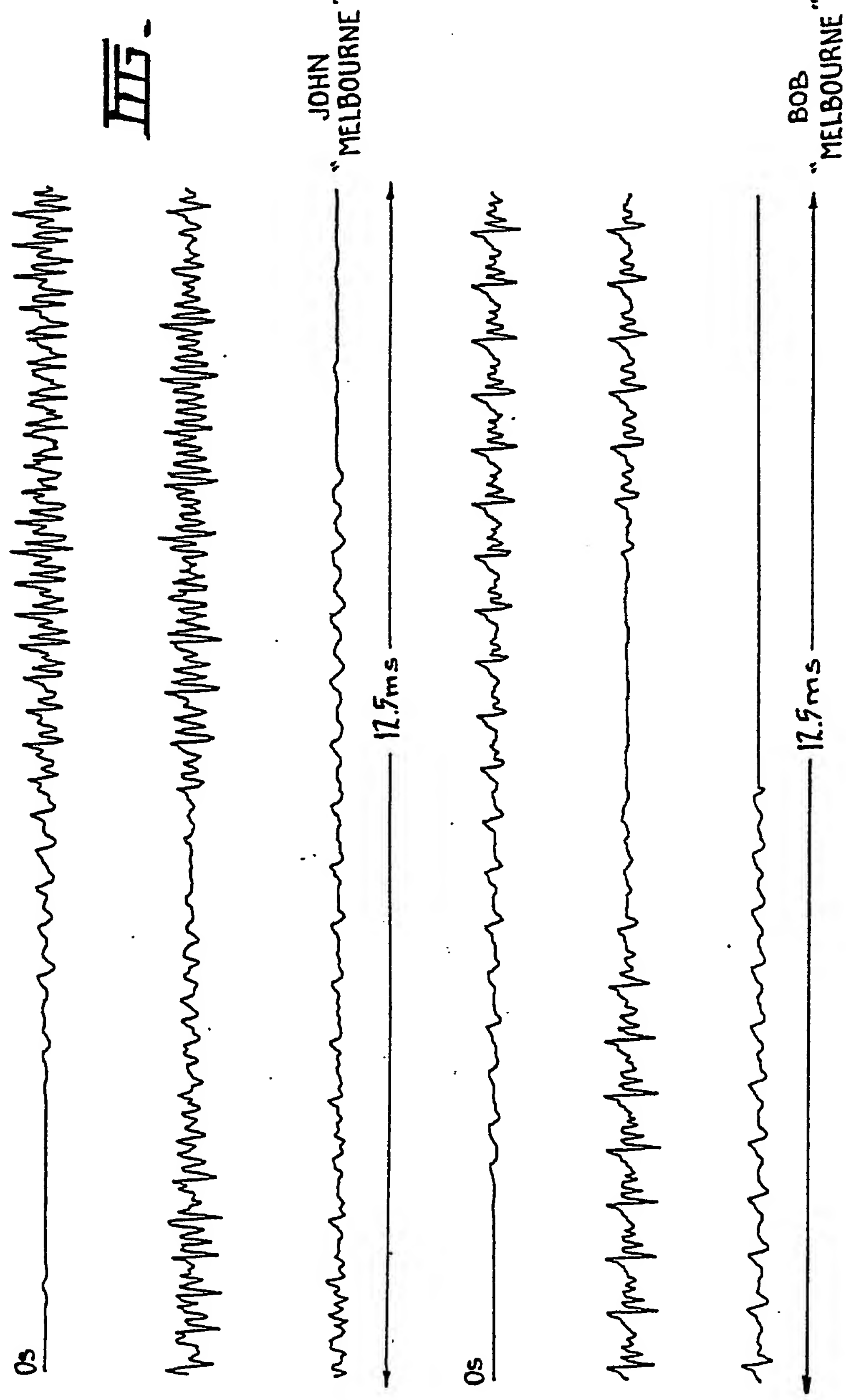


Fig. 1.

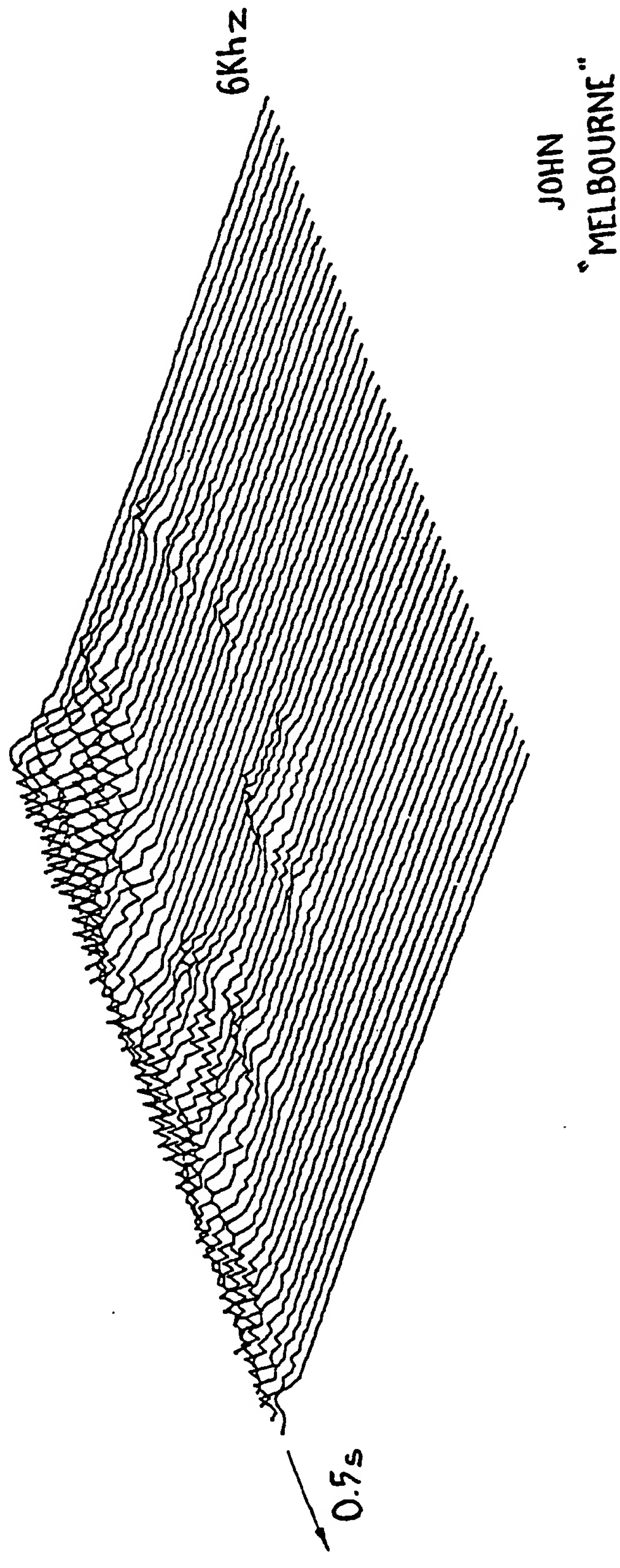
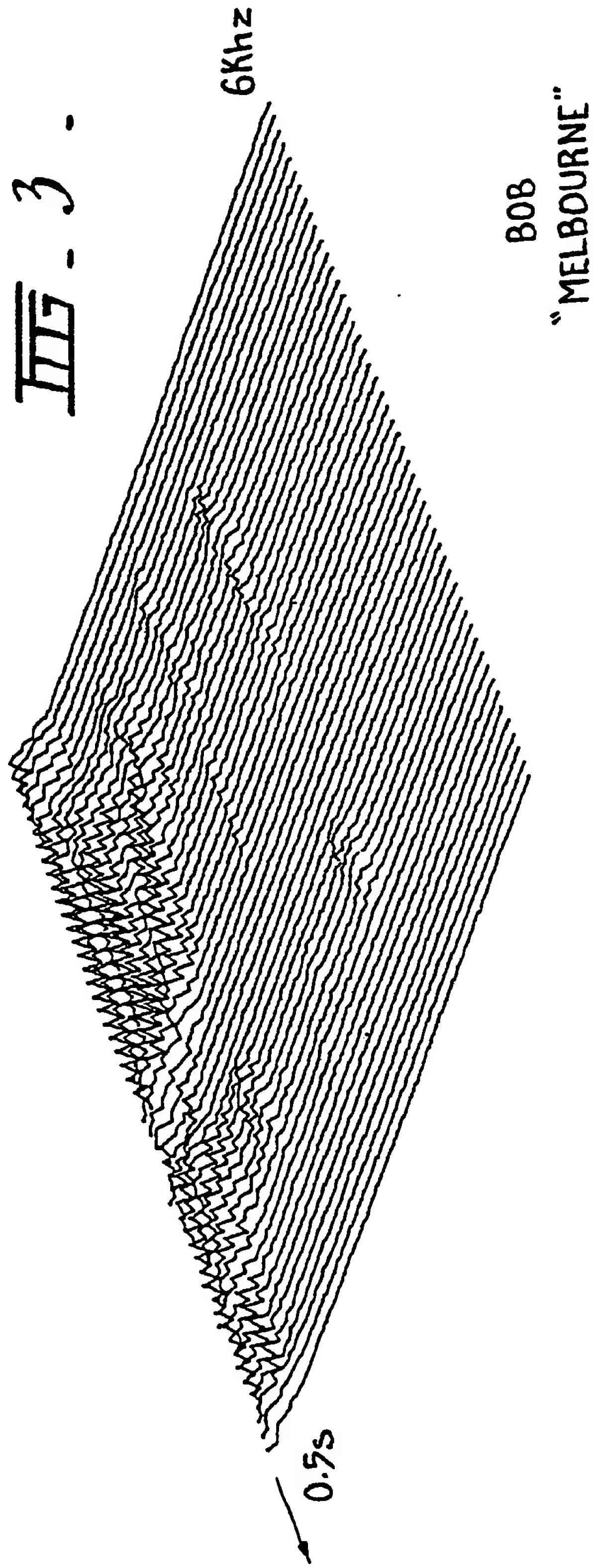


FIG. 2.

FIG. 3



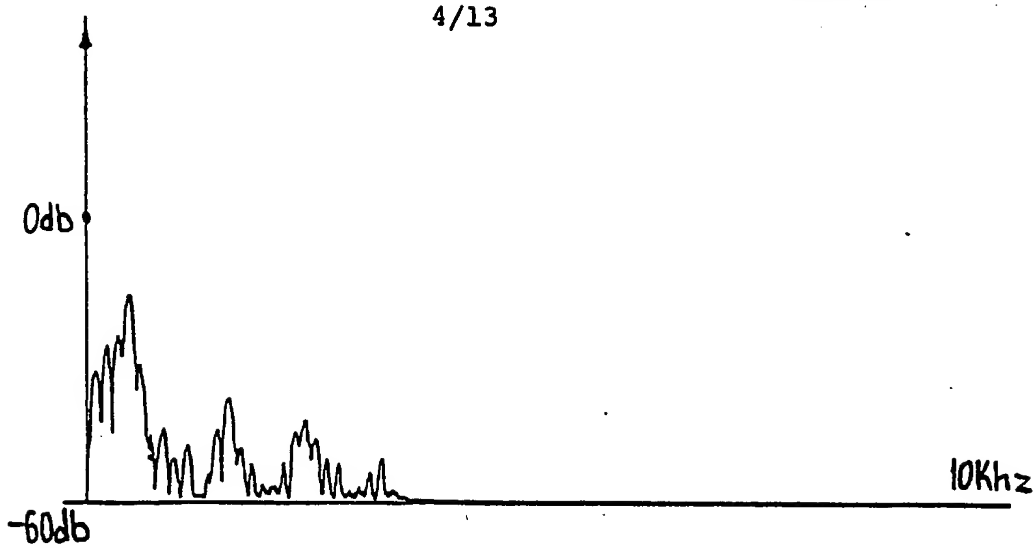
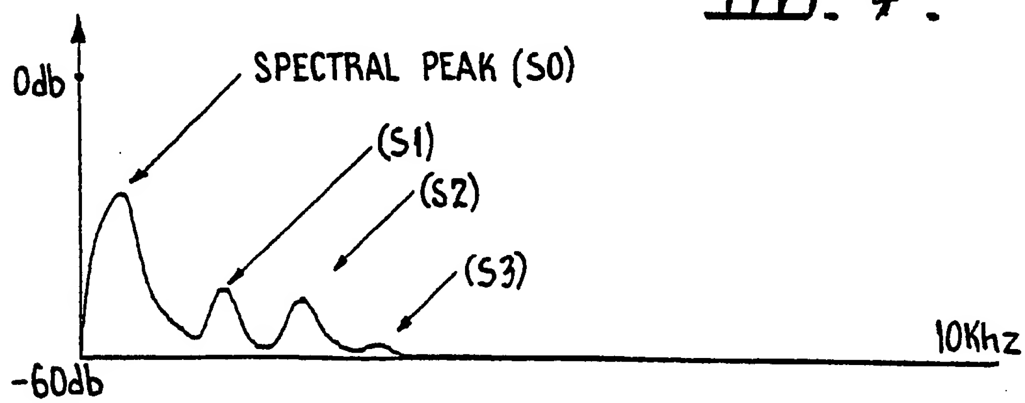


FIG. 4.



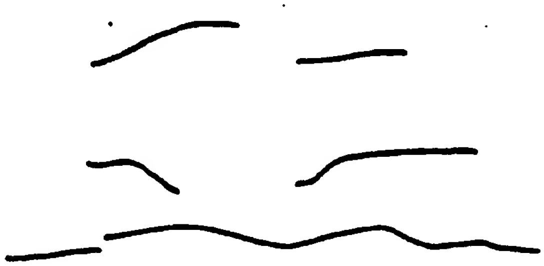


FIG. 17.

FIG. 17A.

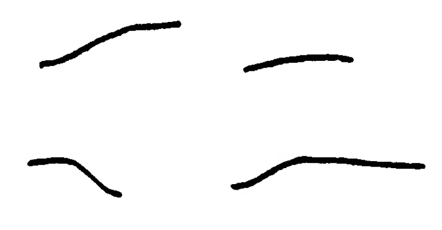


FIG. 5.

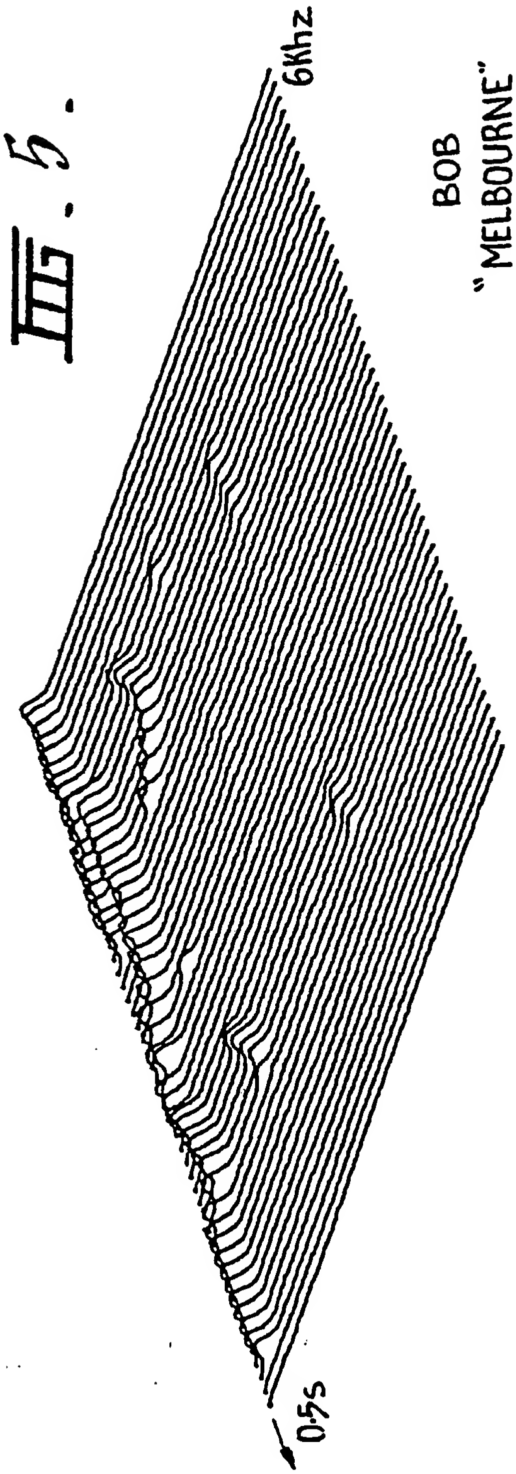
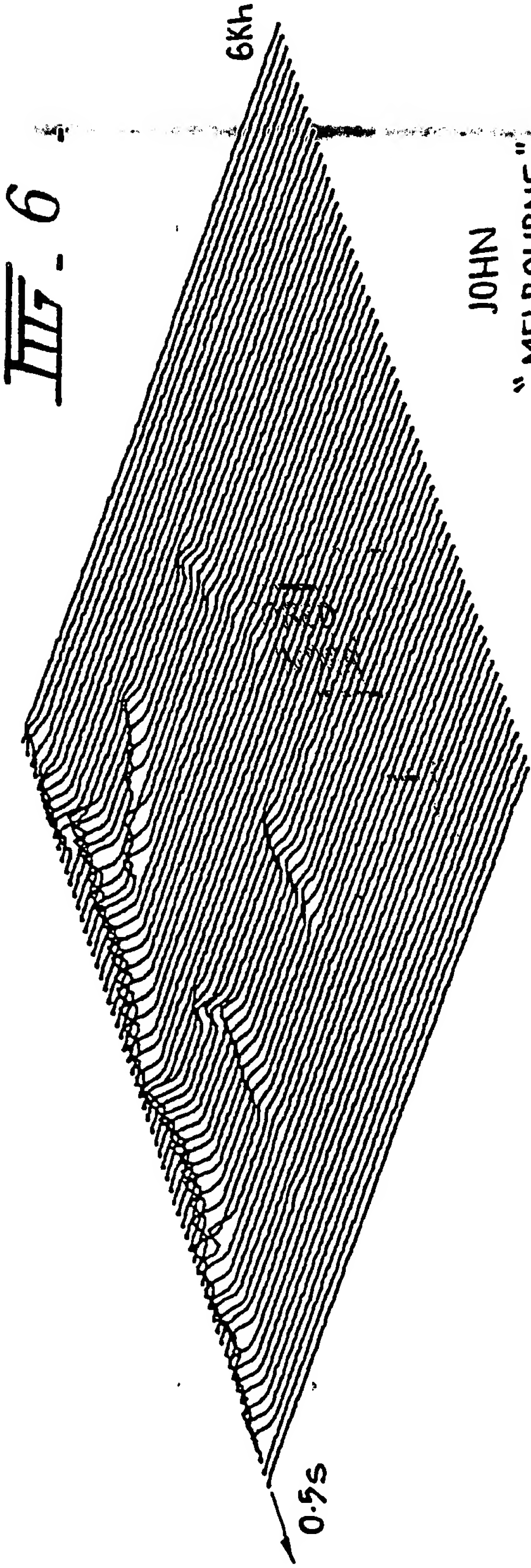


FIG. 6



SYSTEM OVERVIEW

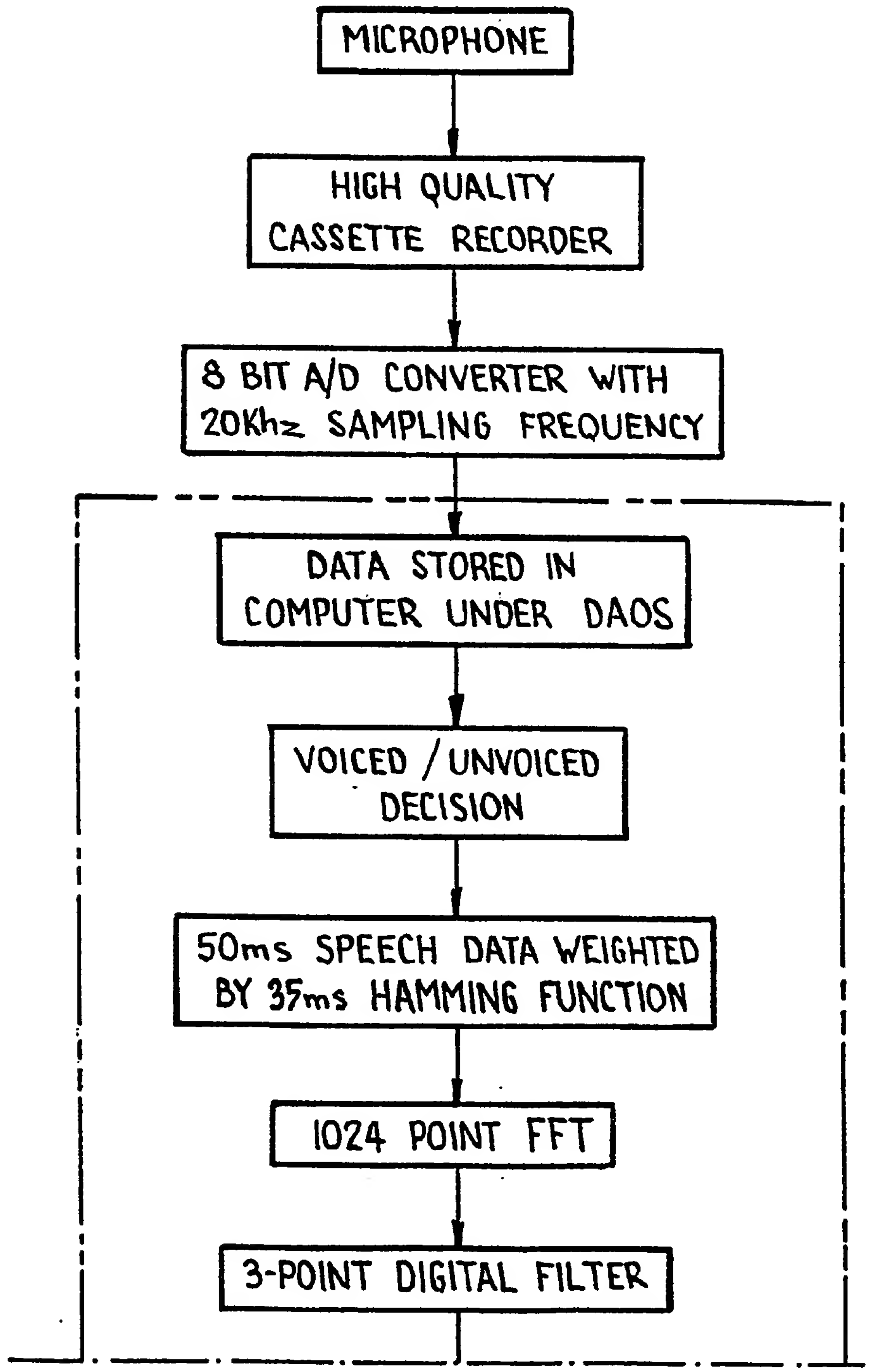


FIG. 7A.

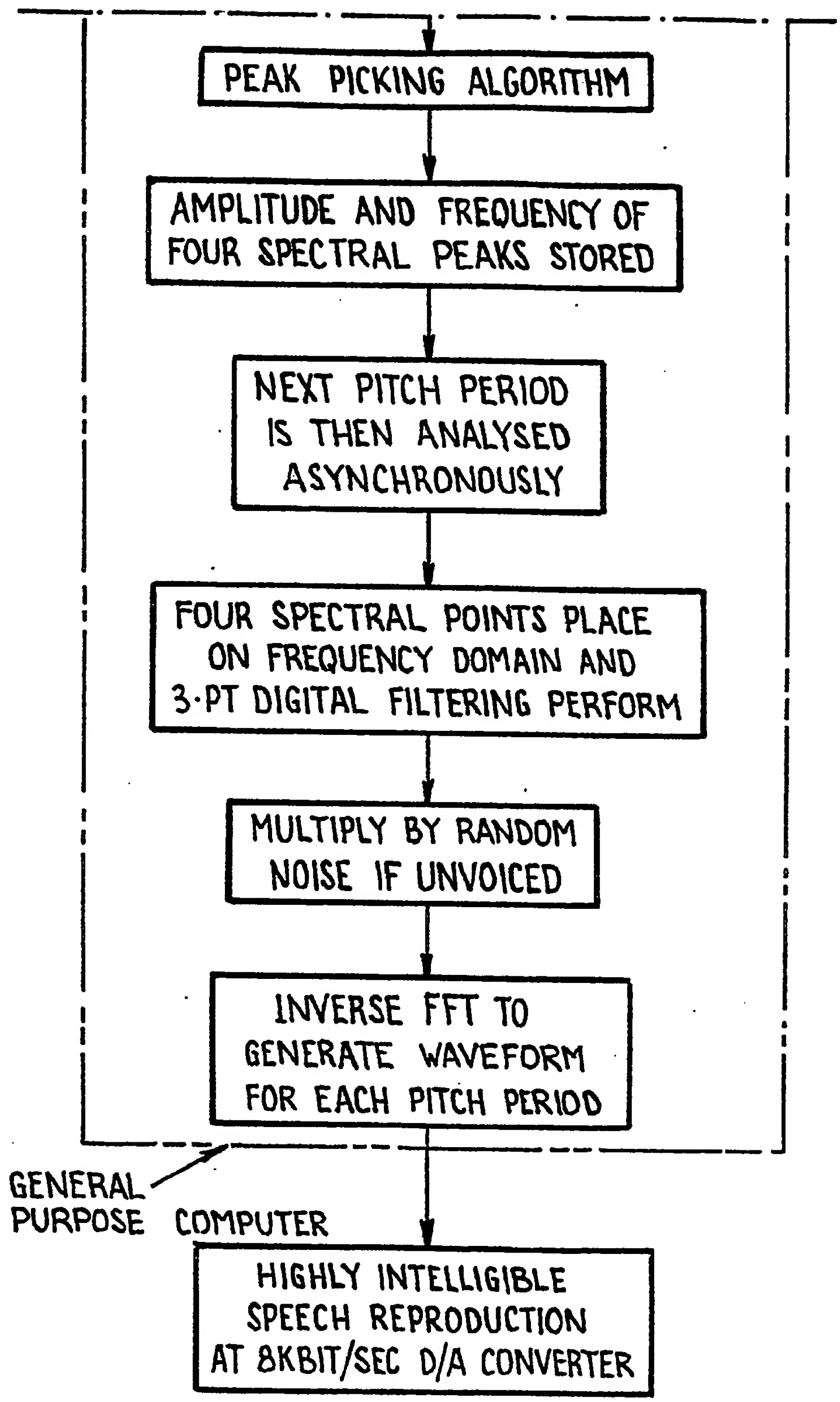

PEAK PICKING ALGORITHM
AMPLITUDE AND FREQUENCY OF FOUR SPECTRAL PEAKS STORED
NEXT PITCH PERIOD IS THEN ANALYSED ASYNCHRONOUSLY
FOUR SPECTRAL POINTS PLACE ON FREQUENCY DOMAIN AND 3·PT DIGITAL FILTERING PERFORM
MULTIPLY BY RANDOM NOISE IF UNVOICED
INVERSE FFT TO GENERATE WAVEFORM FOR EACH PITCH PERIOD
GENERAL PURPOSE COMPUTER
HIGHLY INTELLIGIBLE SPEECH REPRODUCTION AT 8KBIT/SEC D/A CONVERTER

FIG. 7B.

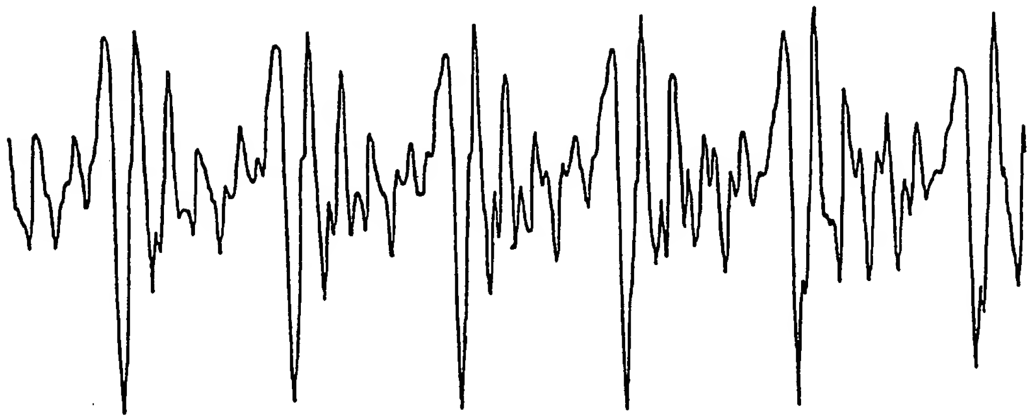


FIG. 8.

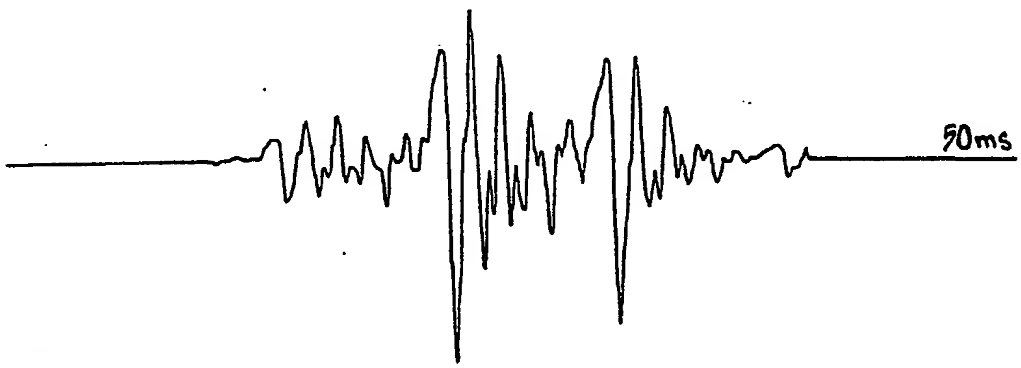


FIG. 9.

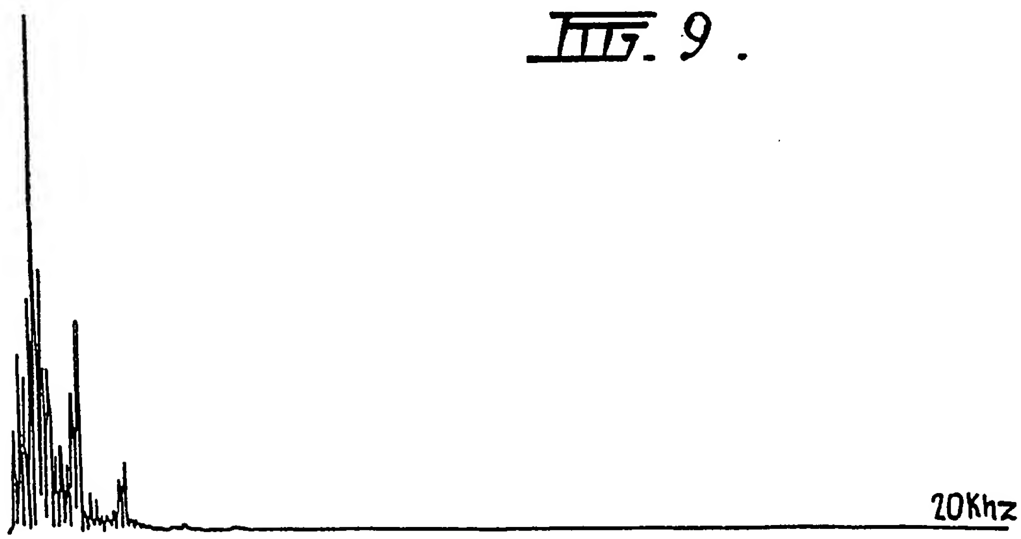


FIG. 10.

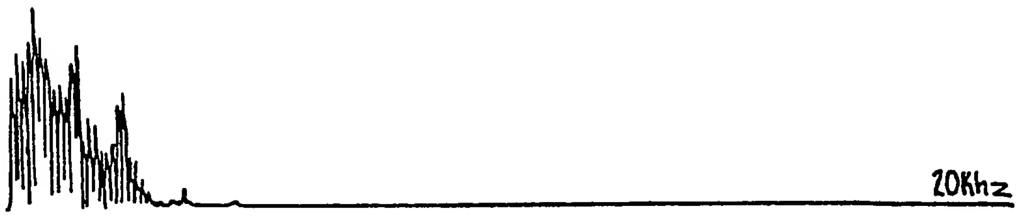


FIG. 11.

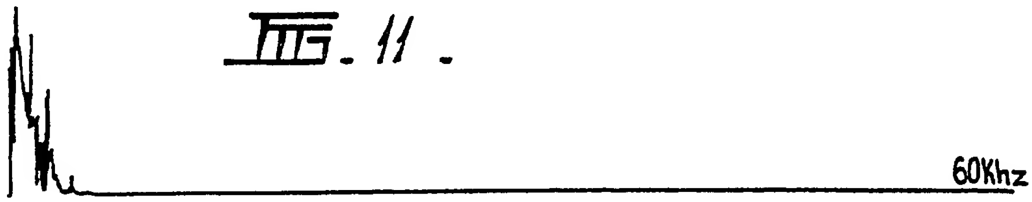


FIG. 12.

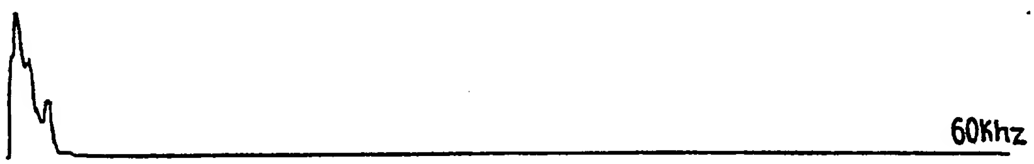


FIG. 13.

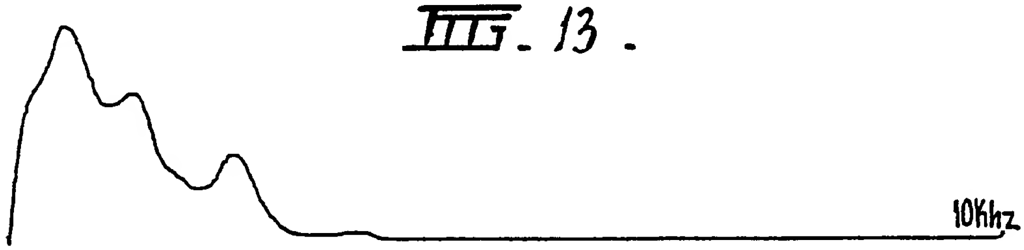


FIG. 14.

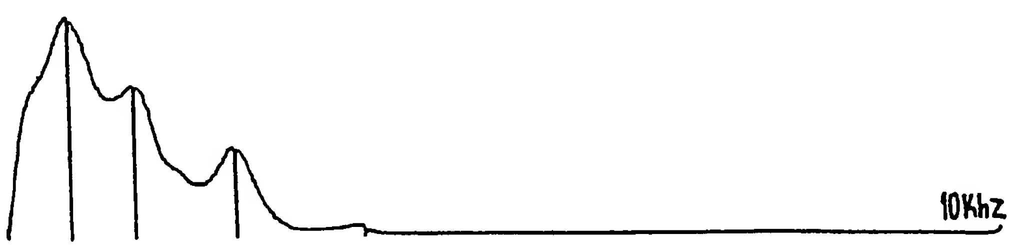
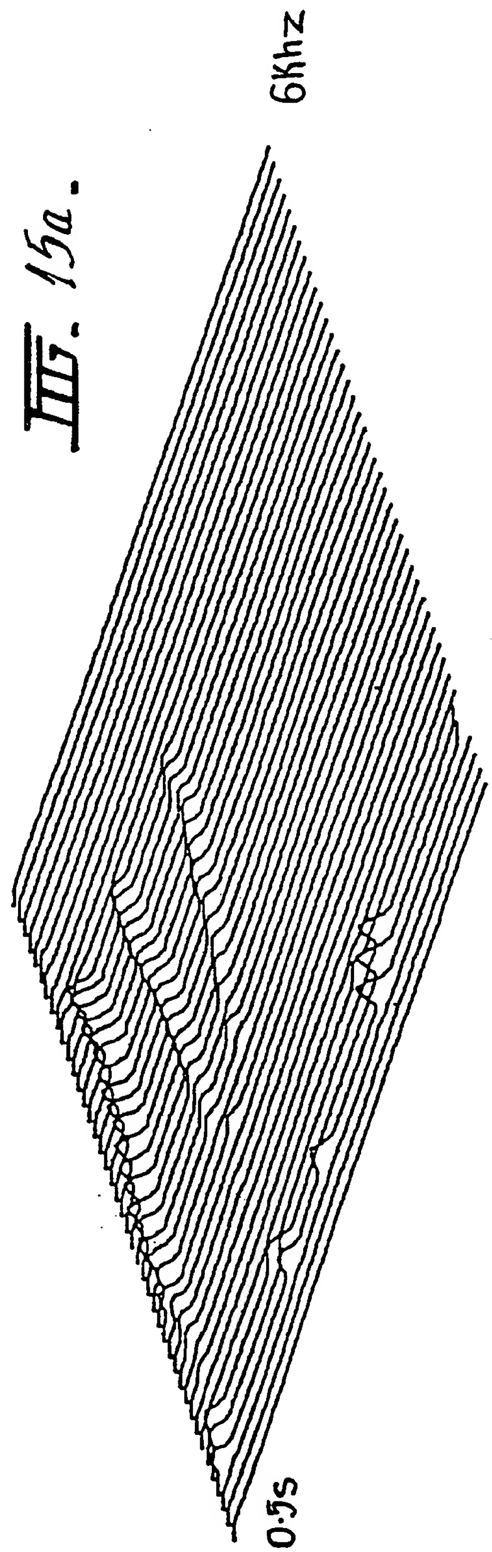


FIG. 15.

FIG. 15a.



REAL TIME SPEECH SYNTHESIS BY DECODING SPECTRAL PEAK INFORMATION.

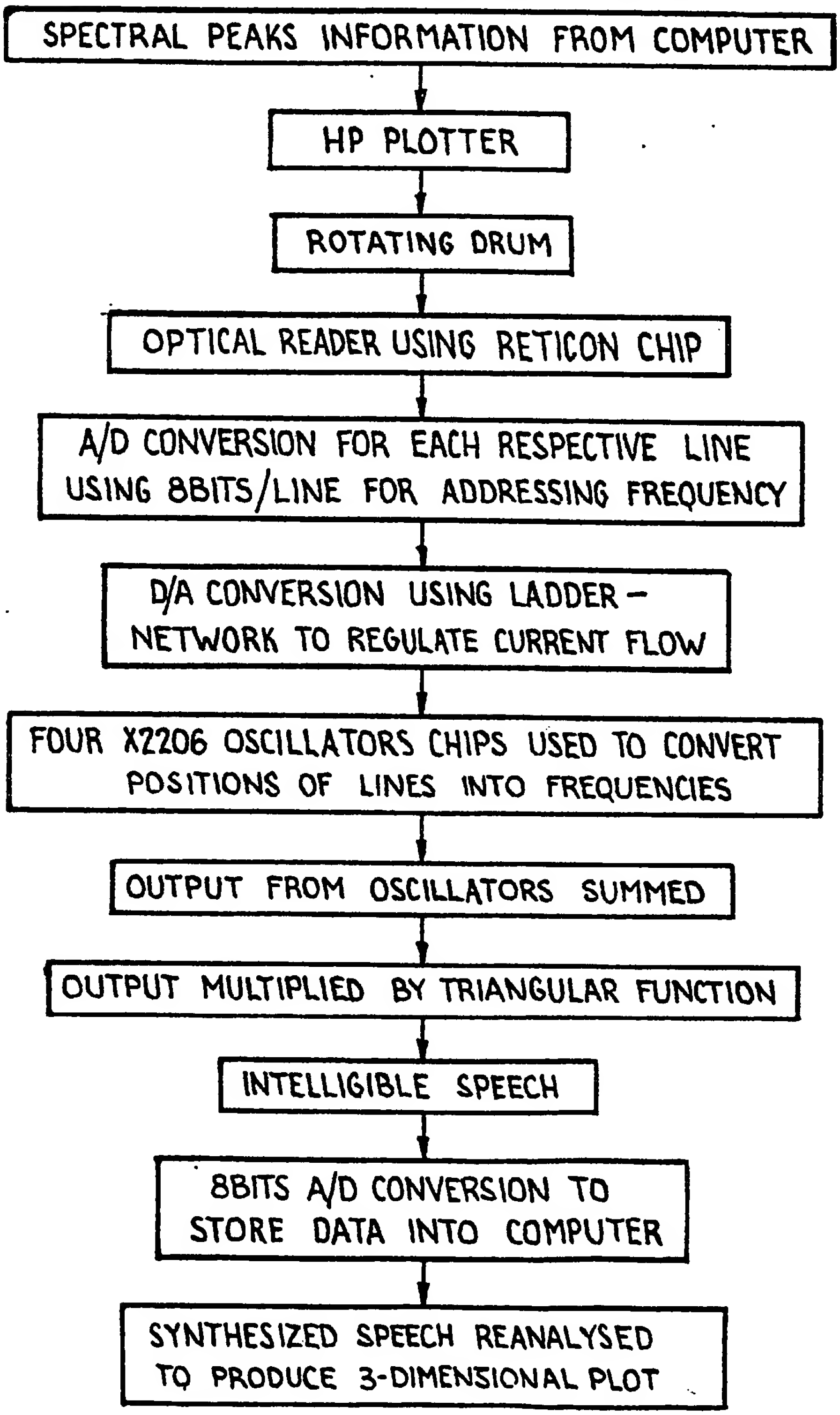


FIG. 16

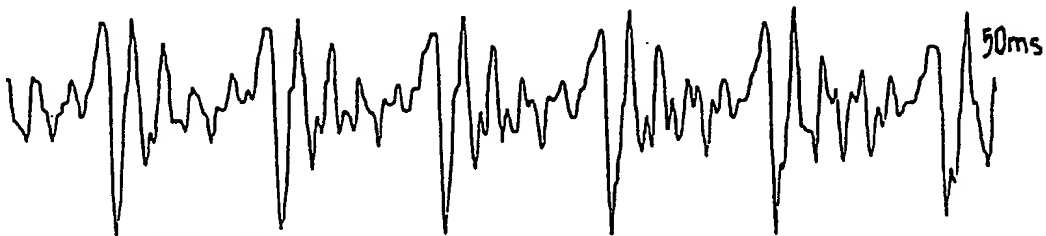


FIG. 18.

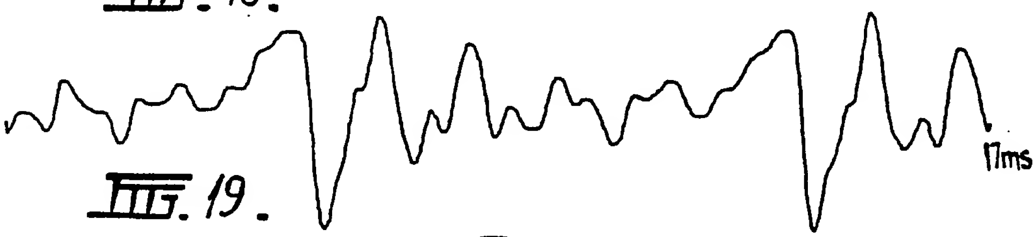


FIG. 19.

FIG. 20.

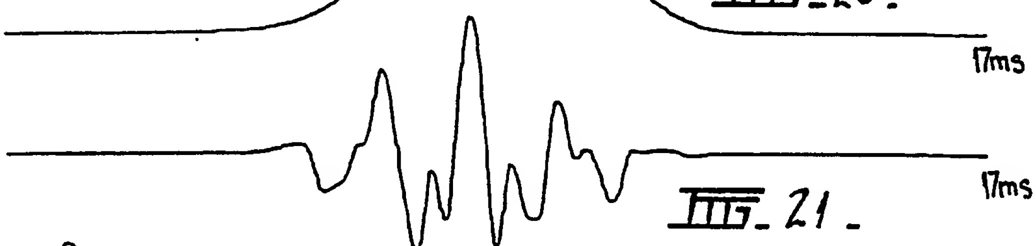


FIG. 21.

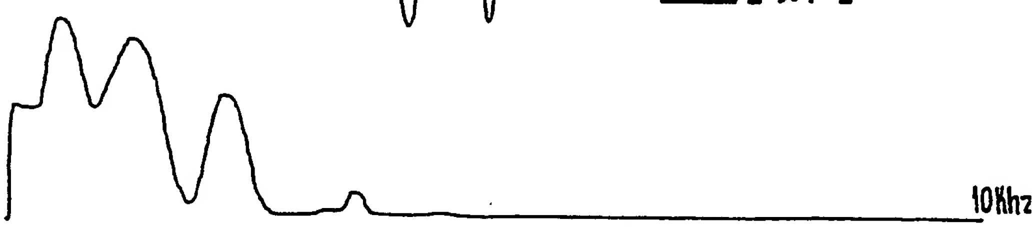


FIG. 22.

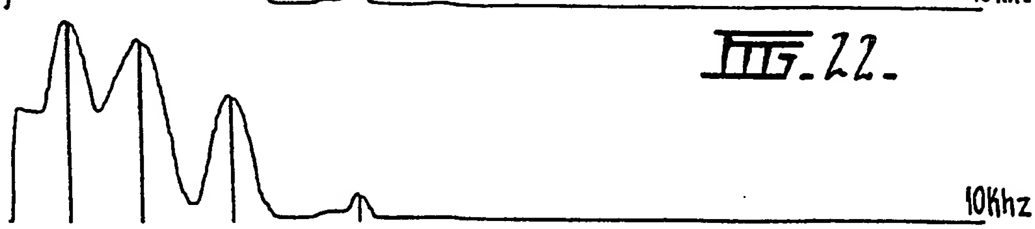


FIG. 23.

European Patent Office

0132216

Application number

EUROPEAN SEARCH REPORT

EP 84 63 0096

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int. Cl. 3)
X	ELECTRONICS AND COMMUNICATIONS IN JAPAN, vol. 62-A, no. 4, 1979, pages 10-17, Scripta Publishing Co., Washington, US; S. IMAI et al.: "Spectral envelope extraction by improved cepstral method" * paragraph 2.1 "Cepstral method" *	1,2,4, 6,12, 15,16	G 10 L 1/02
A	THE JOURNAL OF THE ACOUSTICAL SOCIETY OF JAPAN, vol. 32, no. 1, January 1976, pages 12-23, Tokyo, JP; T. MATSUOKA et al.: "Investigation on phonemic information of static properties of local peaks in the speech spectra" * abstract *	1,15	
			TECHNICAL FIELDS SEARCHED (Int. Cl. 3)
A	IEEE TRANSACTIONS ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, vol. ASSP-22, no. 5, October 1974, pages 362-381, IEEE, New York, US; H.F. SILVERMAN et al.: "A parametrically controlled spectral analysis system for speech" * paragraph V: "Frequency-domain shaping" *	6-11	G 10 L 1/02
A	FR-A-2 337 393 (DIALOG SYSTEMS) * claim 1 *	1-3	

--- -/-

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	12-10-1984	ARMSPACH J.F.A.M.

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document